Transport Layer: TCP/UDP

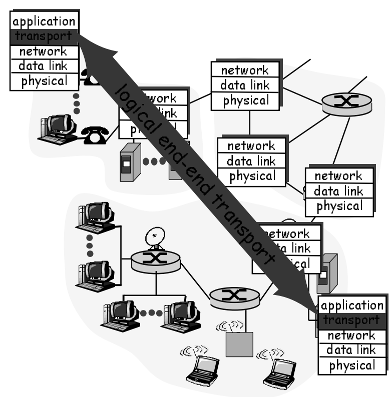
Chapter 24, 16

Transport Layer

- Purpose of transport layer services:
  - multiplexing/demultiplexing
  - reliable data transfer
  - flow control
  - congestion control
- Connection-less transport: UDP
- Connection-oriented transport: TCP
  - reliable transfer
  - flow control
  - connection management

Transport services and protocols

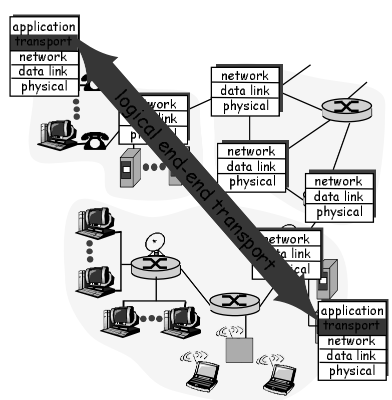
- provide logical communication between application processes running on different hosts
- transport protocols run in end systems via software
- transport vs network layer services:
- network layer: data transfer between end systems
- transport layer: data transfer between processes
  - relies on, enhances, network layer services



Transport-layer protocols

Internet transport services:

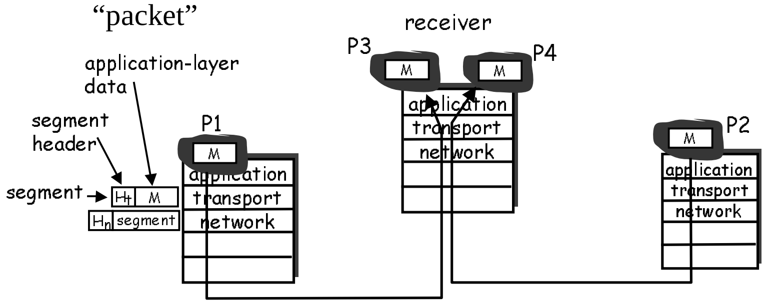
- reliable, in-order unicast delivery (TCP)
  - congestion
  - flow control
  - connection setup
- unreliable (“best-effort”), unordered unicast or multicast delivery: UDP
- services not available:
  - real-time
  - bandwidth guarantees
  - reliable multicast



Multiplexing/demultiplexing

Recall: segment - unit of data exchanged between transport layer entities
- aka TPDU: transport protocol data unit or "packet"

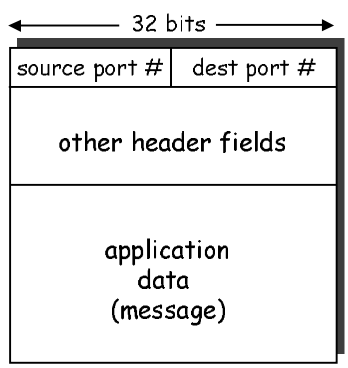
Demultiplexing: delivering received segments to correct app layer processes



Multiplexing/demultiplexing

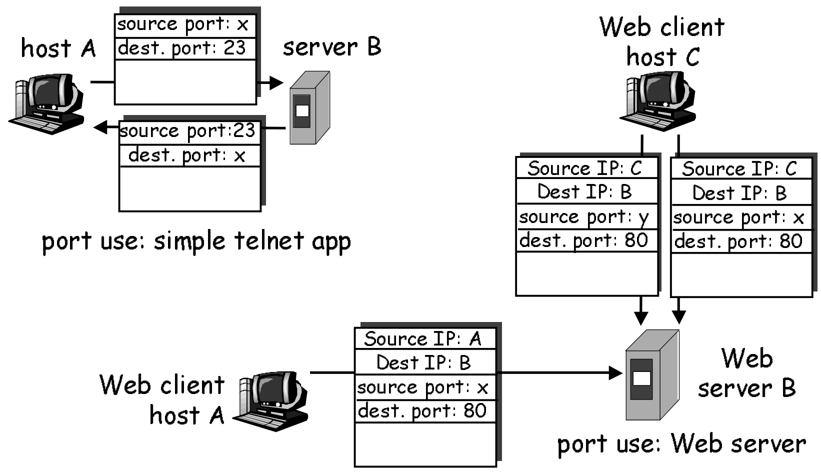
Multiplexing: gathering data from multiple app processes, enveloping data with header (later used for demultiplexing)

multiplexing/demultiplexing:
- based on sender, receiver port numbers, IP addresses
  - source, dest port #s in each segment
  - recall: well-known port numbers for specific applications



TCP/UDP segment format

Multiplexing/demultiplexing: examples



UDP: User Datagram Protocol [RFC 768]

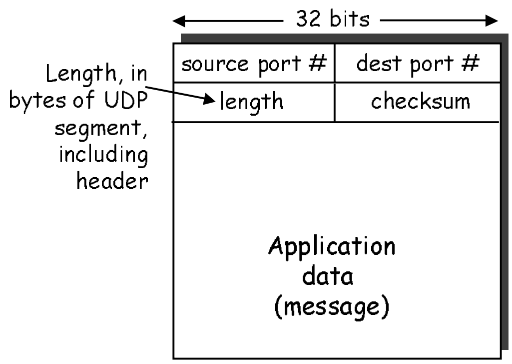
- "no frills," "bare bones" Internet transport protocol
- "best effort" service, UDP segments may be:
  - lost
  - delivered out of order to app
- *connectionless:*
  - no handshaking between UDP sender, receiver
  - each UDP segment handled independently of others

Why is there a UDP?

- no connection establishment (which can add delay)
- simple: no connection state at sender, receiver
- small segment header
- no congestion control: UDP can blast away as fast as desired

UDP: more

- often used for streaming multimedia apps
  - Controversial: no congestion control
- other UDP uses (why?):
  - DNS
  - SNMP
- reliable transfer over UDP: add reliability at application layer
  - application-specific error recover!



UDP segment format

UDP checksum

Goal: detect "errors" (e.g., flipped bits) in transmitted segment

Sender:

- treat segment contents as sequence of 16-bit integers
- checksum: addition (1's complement sum) of segment contents
- sender puts checksum value into UDP checksum field

Receiver:

- compute checksum of received segment
- check if computed checksum equals checksum field value:
  - NO - error detected
    - Toss data OR
    - Pass to app with warning
  - YES - no error detected.

Connection Oriented Transport Protocol Mechanisms

- Properties of connection-oriented Transport Protocols:
  - Logical connection
  - Establishment
  - Maintenance termination
  - Reliable
  - e.g. TCP

Connection-Oriented Transport via Reliable Network Layer

- Transport Layer Services like TCP are complicated – to start, let's first assume we are working with a reliable network layer service
  - e.g. reliable packet switched network using X.25
  - e.g. frame relay using LAPF control protocol
  - e.g. IEEE 802.3 using connection oriented LLC service
  - NOT IP! IP is unreliable
- Assume arbitrary length message
- Transport service is end to end protocol between two systems on same network

Issues in a Simple Transprot Protocol

- If we have a reliable network layer, then the transport layer must consider:
  - Addressing
  - Multiplexing
  - Flow Control
  - Connection establishment and termination

Addressing

- Target user specified by:
  - User identification
    - Usually host, port
      - Called a socket in TCP/UDP
    - Port represents a particular transport service (TS), e.g. HTTPD
  - Transport protocol identification
    - Generally only one per host
    - If more than one, then usually one of each type
      - Specify transport protocol (TCP, UDP)
  - Host address
    - An attached network device
    - In an internet, a global internet address (IP Address)
    - A well-known address or lookup via name server

Multiplexing

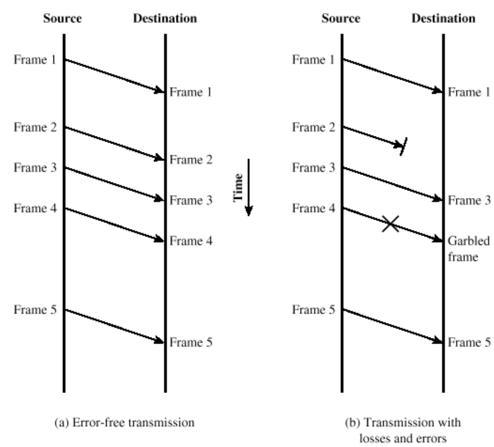
- Multiple users employ same transport protocol
- User identified by port number or service access point (SAP)
- Described previously

Flow Control

- Can be difficult than flow control at the data link layer – data is likely traveling across many networks, not one network. Some potential problems:
  – Longer transmission delay between transport entities compared with actual transmission time
    - Delay in communication of flow control info
  – Variable transmission delay
    - Difficult to use timeouts
- Flow may be controlled because:
  – The receiving user cannot keep up
  – The receiving transport entity cannot keep up
  – If either happens, the results is a buffer that can get full and eventually lose data

Model of Frame Transmission

Diagram for Frame/Packet Transmission



We’ll use this model to discuss flow control issues

Coping with Flow Control Requirements (1)

- Do nothing
  - Segments that overflow are discarded
  - Sending transport entity will fail to get ACK and will retransmit
    - Thus further adding to incoming data and could exacerbate the flow control problem
- Refuse further segments from network layer
  - Clumsy
  - Multiplexed connections are controlled on aggregate flow

Coping with Flow Control Requirements (2)

- One protocol: Stop-and-Wait
  - Sender must wait for recipient to send ACK before sending the next packet
    - Not very efficient usage of the network, only one outstanding message can be in transit at a time
  - Works well on reliable network
    - Failure to receive ACK is taken as flow control indication
  - Does not work well on unreliable network
    - Cannot distinguish between lost segment and flow control

Coping with Flow Control

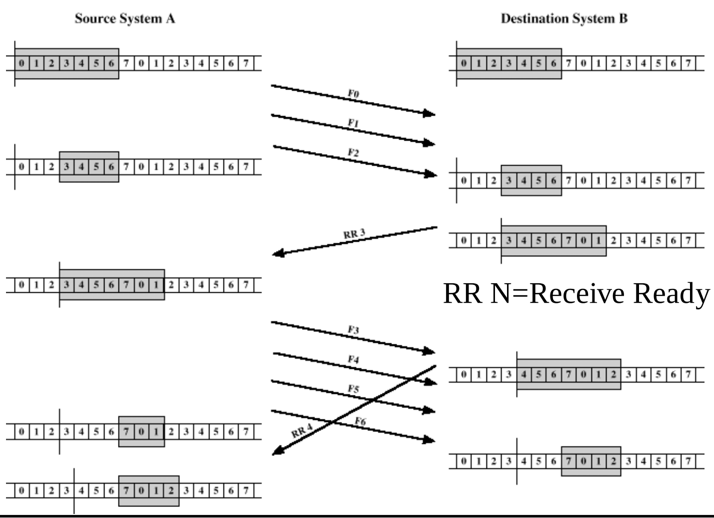
- Credit-Based Scheme
  - Credit = How much data sender can transmit
    - Sliding window idea, sender can send a number of frames up to the window size
    - Receiver sends single ACK that acknowledges all previous frames
    - Window size varies based on credit available
    - Receiver can control credit of the sender
      - In acknowledgement, receiver could change the window size
  - Advantages
    - Better network usage, allows outstanding messages to be in transit than Stop-And-Wait
    - More effective on unreliable network
      - Decouples flow control from ACK
    - May ACK without granting credit and vice versa
  - Each octet has sequence number
  - Each transport segment has a sequence number, acknowledgement number and window size in header

Sliding Window Enhancements

- Receiver can acknowledge frames without permitting further transmission (Receive Not Ready)
- Must send a normal acknowledge to resume
- If full duplex two-way communications, we need two windows: one for transmit and one for receive
  - Piggybacking – if sending data and acknowledgement frame, combine together
- More efficient than stop-and-wait since many frames may be in the pipeline

Example Sliding Window

Fixed Window Size



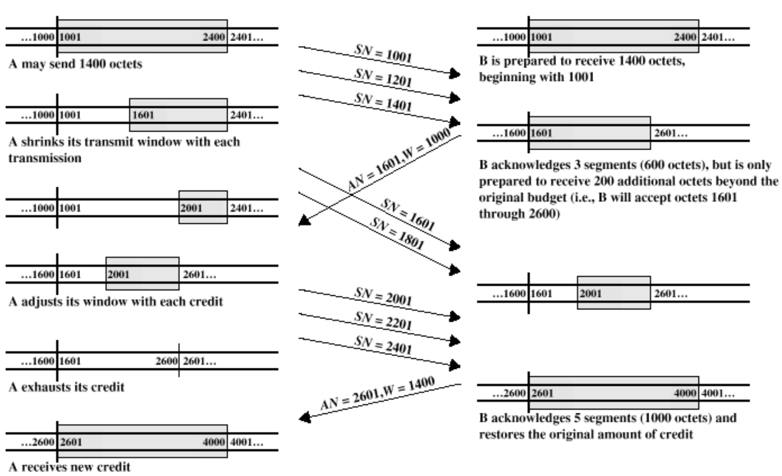
RR N=Receive Ready on N

Use of Header Fields

- For credit-based window size
  - When sending, Sequence Number is that of first octet in segment
  - ACK includes AN=i (Acknowledgement Number), W=j (Window Size)
  - All octets through SN=i-1 acknowledged
    - Next expected octet is i
  - Permission to send additional window of W=j octets
    - i.e. octets through i+j-1

Credit Allocation Example

A, B initially “synched”

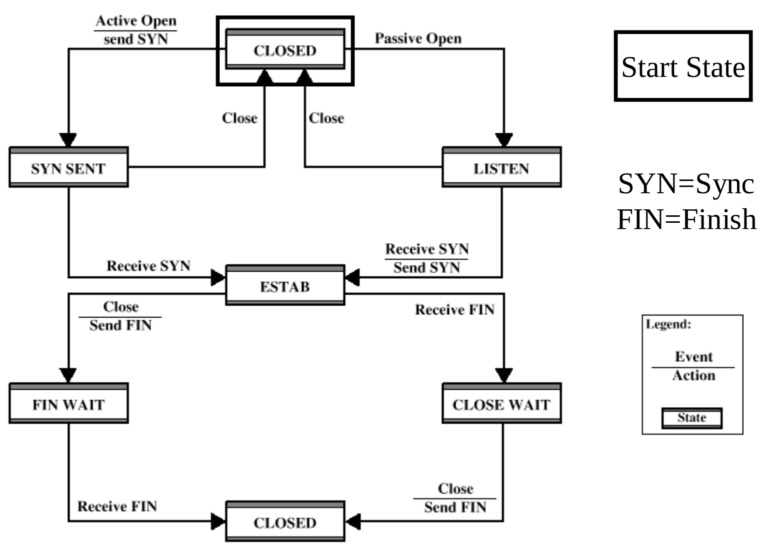


200 octets per segment

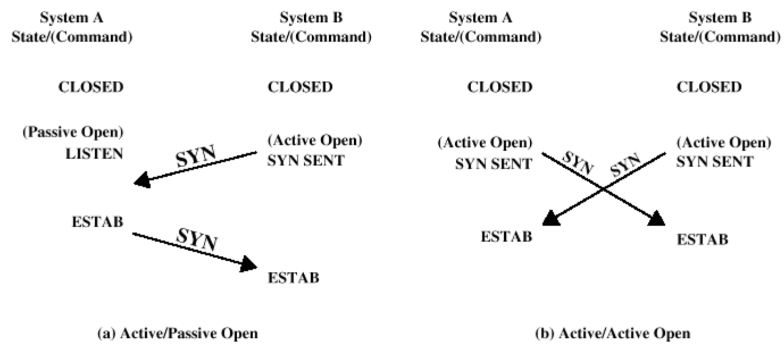
Establishment and Termination

- Even with a reliable network service, both ends need to "set up" the connection:
  - Allow each end to know the other exists and is listening
  - Negotiation of optional parameters
    - Maximum Segment Size
    - Maximum Window Size
  - Triggers allocation of transport entity resources
    - Buffer space allocated
    - Entry in connection tables

Connection State Diagram – Reliable Network Service



Connection Establishment



Setting up the connection

- What if a SYN received while not in the Listen state?
  - Reject with RST (Reset)
  - Queue request until matching open issued
  - Signal TS user to notify of pending request

Termination

- Connection can be terminated by sending FIN
- Graceful termination
  - CLOSE_WAIT state and FIN_WAIT must accept incoming data until FIN received
  - Ensures both sides have received all outstanding data and that both sides agree to connection termination before actual termination

Unreliable Network Service

- Now let's look at the more general case if we are building our transport service on top of an unreliable network layer

- An unreliable network service makes the transport layer much more complicated if we want to ensure reliability
- Examples of unreliable network services:
  - Internet using IP,
  - Frame Relay using LAPF
  - IEEE 802.3 using unacknowledged connectionless LLC
- Segments may get lost
- Segments may arrive out of order

Problems

- Ordered Delivery
- Retransmission strategy
- Duplication detection
- Flow control
- Connection establishment
- Connection termination
- Crash recovery

Ordered Delivery

- Segments may arrive out of order
- Number segments sequentially
- TCP numbers each octet sequentially
- Segments are numbered by the first octet number in the segment

- TCP actually numbers segments starting at a random value!
  – Minimizes possibility that a segment still in the network from an earlier, terminated connection between the same hosts is mistaken for a valid segment in a later connection (who would also have to happen to use the same port numbers)

Retransmission Strategy

- Need to re-transmit when
  - Segment damaged in transit
  - Segment fails to arrive
- Receiver must acknowledge successful receipt
- Use cumulative acknowledgement
- Time out waiting for ACK triggers re-transmission

- How long to wait until re-transmitting?
  - Too short: duplicate data
  - Too long: Unnecessary delay delivering data

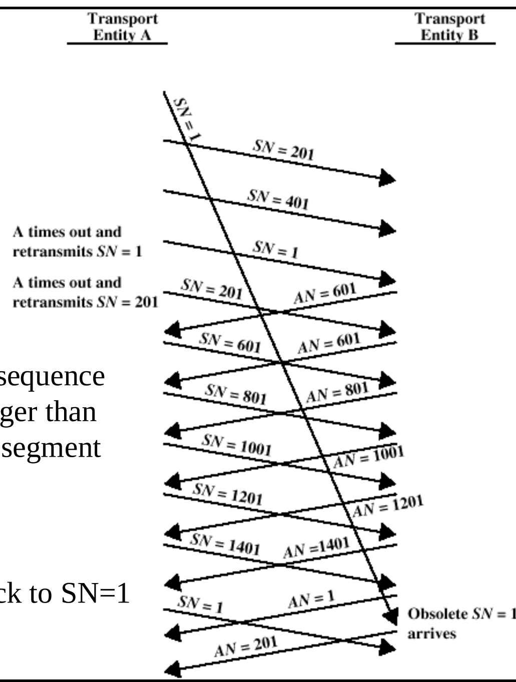
Timer Value

- Fixed timer
  - Based on understanding of network behavior
  - Can not adapt to changing network conditions
  - Too small leads to unnecessary re-transmissions
  - Too large and response to lost segments is slow
  - Should be a bit longer than Round Trip Time (RTT)
- Adaptive scheme
  - E.g. set timer to average of previous ACKs
  - Problems:
    - Sender may not ACK immediately
    - Cannot distinguish between ACK of original segment and re-transmitted segment
    - Conditions may change suddenly

Duplication Detection

- If ACK lost, segment is re-transmitted
- Receiver must recognize duplicates
- Duplicate received prior to closing connection
  - Receiver assumes ACK lost and ACKs duplicate
  - Sender must not get confused with multiple ACKs
  - Sequence number space large enough to not cycle within maximum life of segment

- Also possible to receive a duplicate after closing the connection!



Illustrates need for the sequence number space to be larger than the maximum possible segment lifetime

Note: cycle back to SN=1

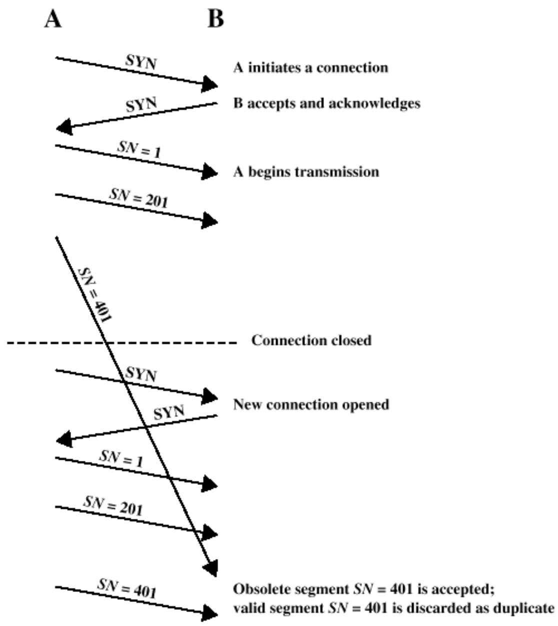
Flow Control

- Can use credit allocation described earlier
- Generally little harm if a single ACK/Credit segment is lost, will resynchronize the next time
- Problem if B sends AN=i, W=0 closing window
- Later, B sends AN=i, W=j to reopen, but this is lost
- Sender thinks window is closed, receiver thinks it is open
- Solution: use window timer
- If timer expires, send something to break the deadlock
  - Could be re-transmission of previous segment

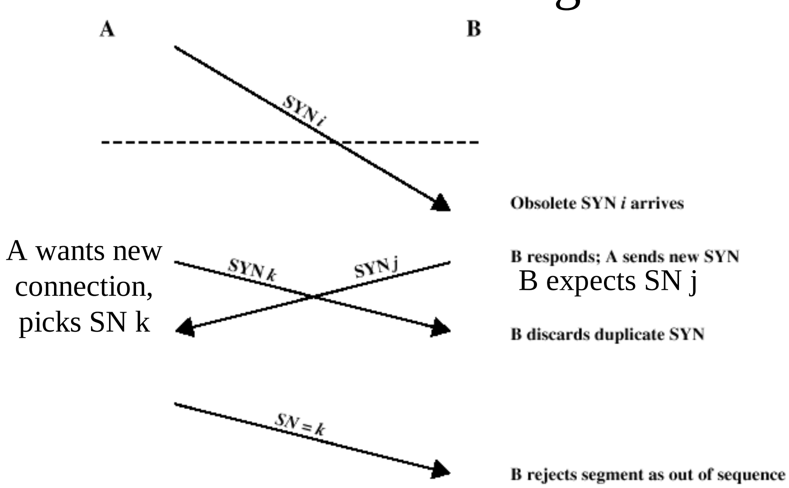
Connection Establishment

- Two way handshake
  - A send SYN, B replies with SYN
  - Lost SYN handled by re-transmission
    - Can lead to duplicate SYNs
  - Ignore duplicate SYNs once connected
- Lost or delayed data segments can cause connection problems
  - Segment from old connections

Two Way Handshake: Obsolete Data Segment



Two Way Handshake: Obsolete SYN Segment

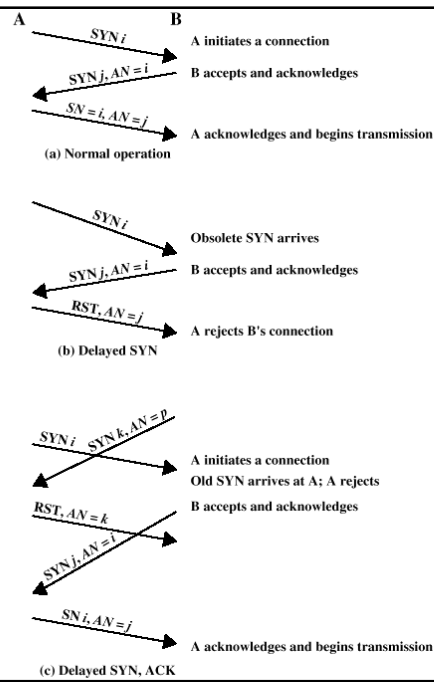


Connection Establishment – Three Way Handshake

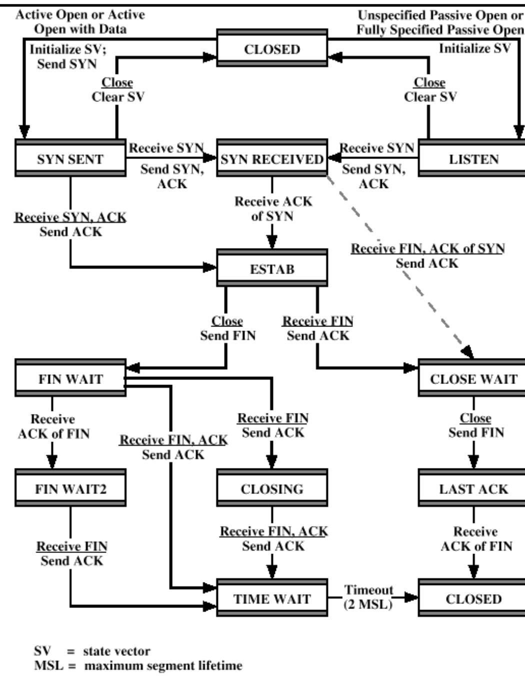
- Solution: Explicitly acknowledge each other's SYN and sequence number
  - Use SYN i
  - Need ACK to include i

- Called the Three Way Handshake

Three Way Handshake: Examples



Three Way Handshake: State Diagram



Connection Termination

- Same problems we had with connection establishment can also occur with connection termination
  - Lost or obsolete FIN segment
  - Can lose last data segment if FIN arrives before last data segment
- Solution: associate sequence number with FIN
- Receiver waits for all segments before FIN sequence number
- Must explicitly ACK FIN

Graceful Close

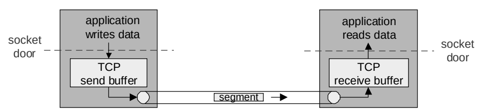
- Send FIN i and receive AN i
- Receive FIN j and send AN j
- Wait twice maximum expected segment lifetime

Crash Recovery

- If the transport service crashes and restarts, after restart all state info is lost
- Connection is half open
  - Side that did not crash still thinks it is connected
- Close connection using persistence timer
  - Wait for ACK for (time out) * (number of retries)
  - When expired, close connection and inform user
- Send RST i in response to any i segment arriving
- User must decide whether to reconnect
  - Problems with lost or duplicate data

TCP:Overview RFCs: 793, 1122, 1323, 2018, 2581

- point-to-point:
  - one sender, one receiver
- reliable, in-order *byte stream:*
  - no "message boundaries"
- pipelined:
  - TCP congestion and flow control set window size
- send & receive buffers

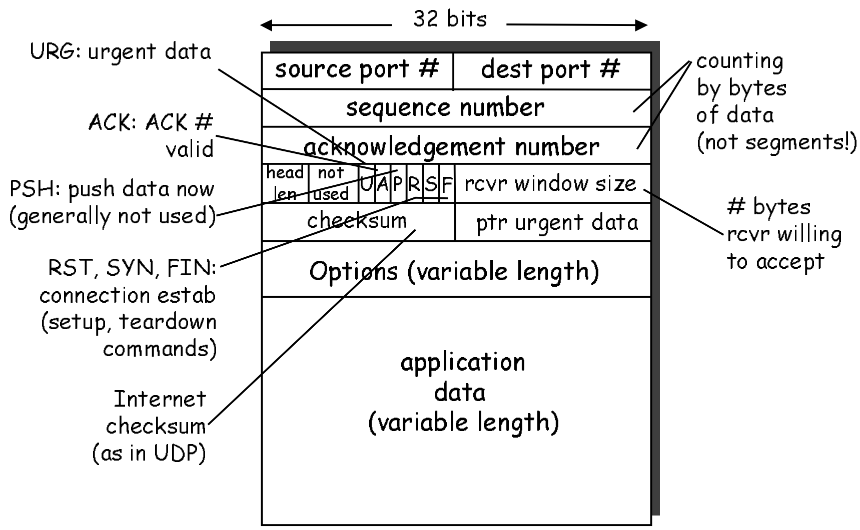


- full duplex data:
  - bi-directional data flow in same connection
  - MSS: maximum segment size
- connection-oriented:
  - handshaking (exchange of control msgs) init's sender, receiver state before data exchange
- flow controlled:
  - sender will not overwhelm receiver

TCP Properties

- stream orientation. stream of OCTETS (bytes) passed between send/ recv
- byte stream is full duplex
  - think of it as two independent streams joined with a **piggybacking** mechanism
  - piggybacking - one data stream has control info for the other data stream (going the other way)
- unstructured stream
  - TCP doesn't show packet boundaries to applications
  - But you can still structure your message if you want
  - Recall usage with sockets:
    - One write() call to send data
    - May require multiple read() calls

TCP segment structure



TCP Fields

- Source, Destination Port: 16 bits each
- Sequence Number: 32 bits
  - Sequence # of first data octet in the segment, initialized randomly as described earlier
- ACK Number: 32 bits
  - Piggybacked ACK, contains sequence number of the next data octet the receiver expects
- Header Len: 4 bits
  - Number of 32 bit words in the header
- Not Used: 6 bits for future use

TCP Fields

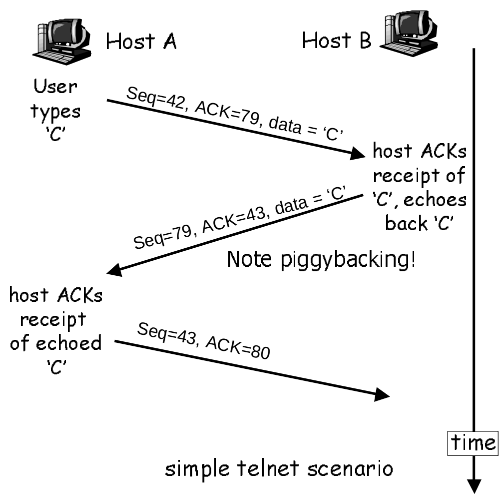
- Flags – 6 bits
  - URG – Urgent Pointer field significant
  - ACK – Ack field significant
  - PSH – Push (flush or "push" buffer now, send data to app)
  - RST – Reset connection
  - SYN – Synchronize sequence numbers
  - FIN – No more data
- Window – 16 bits
  - Flow control credit allocation
- Checksum – 16 bits
  - One's complement sum as in UDP
- Urgent Pointer – 16 bits
  - Last octet in a seq of "urgent" data. Sometimes not interpreted. Urgent data should be processed now, even before any data sitting in the buffer (e.g. send control-c to terminate)
- Options – Variable
  - Support for timestamping, negotiating MSS

TCP seq. #'s and ACKs

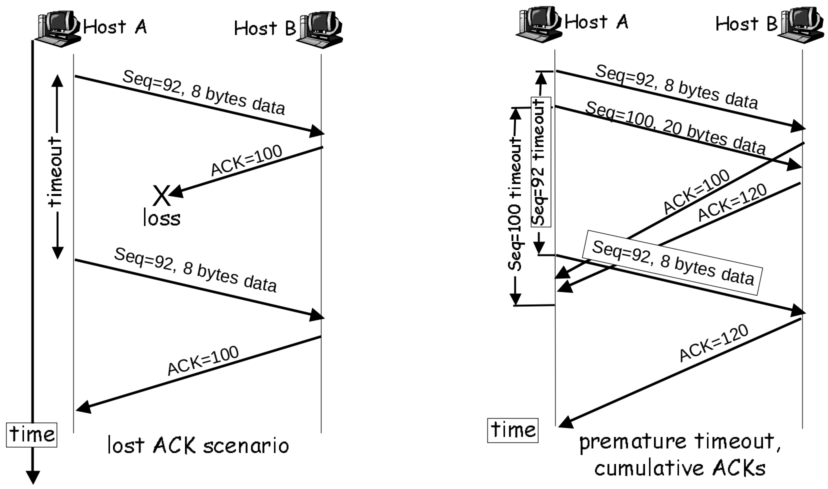
<u>Seq. #'s:</u>
- byte stream "number" of first byte in segment's data

<u>ACKs:</u>
- seq # of next byte expected from other side
- cumulative ACK

Q: How does the receiver handles out-of-order segments?
- A: TCP spec doesn't say, - up to implementer



TCP: retransmission scenarios



Sender must be smart enough to ignore duplicate ACK

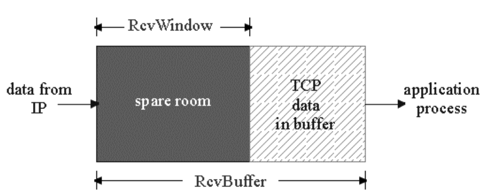
TCP Flow Control

flow control

sender won't overrun receiver's buffers by transmitting too much, too fast

`RcvBuffer` = size or TCP Receive Buffer

`RcvWindow` = amount of spare room in Buffer



receiver buffering

receiver: explicitly informs sender of (dynamically changing) amount of free buffer space

– **`RcvWindow` field** in TCP segment

sender: keeps the amount of transmitted, unACKed data less than most recently received **`RcvWindow`**

TCP Round Trip Time and Timeout

Q: how to set TCP timeout value?

- longer than RTT
  - note: RTT will vary
- too short: premature timeout
  - unnecessary retransmissions
- too long: slow reaction to segment loss

Q: how to estimate RTT?

- **`SampleRTT:`** measured time from segment transmission until ACK receipt
  - ignore retransmissions, cumulatively ACKed segments
- **`SampleRTT`** will vary, want estimated RTT "smoother"
  - use several recent measurements, not just current **`SampleRTT`**

TCP Round Trip Time and Timeout

```
EstimatedRTT = (1-x)*EstimatedRTT + x*SampleRTT
```

- Exponential weighted moving average
- influence of given sample decreases exponentially fast
- typical value of x: 0.1

Setting the timeout

- **`EstimatedRTT`** plus "safety margin"
- large variation in **`EstimatedRTT ->`** larger safety margin

```
Timeout = EstimatedRTT + 4*Deviation
```

TCP Connection Management

Three way handshake:

Step 1: client end system sends TCP SYN control segment to server
– specifies initial seq #

Step 2: server end system receives SYN, replies with SYNACK control segment
– ACKs received SYN
– allocates buffers
– specifies server-> receiver initial seq. #

TCP Connection Management (cont.)

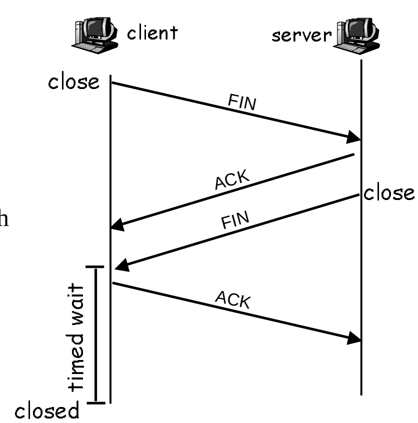
Closing a connection:

Step 1: client end system sends TCP FIN control segment to server

Step 2: server receives FIN, replies with ACK. Closes connection, sends FIN.

Step 3: client receives FIN, replies with ACK.
– Enters “timed wait” - will respond with ACK to received FINs

Step 4: server, receives ACK. Connection closed.



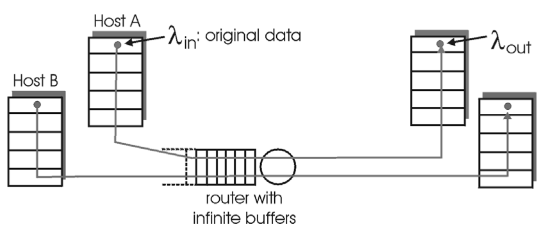
Principles of Congestion Control

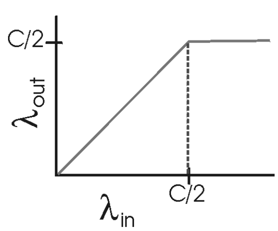
Congestion:

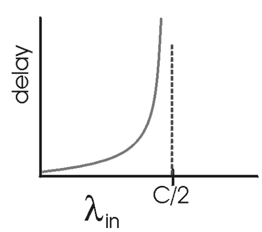
- informally: “too many sources sending too much data too fast for *network* to handle”
- different from flow control!
- manifestations:
  – lost packets (buffer overflow at routers)
  – long delays (queueing in router buffers)
- A top-10 problem!

Causes/costs of congestion: scenario 1

- two senders, two receivers
- one router, infinite buffers
- no retransmission



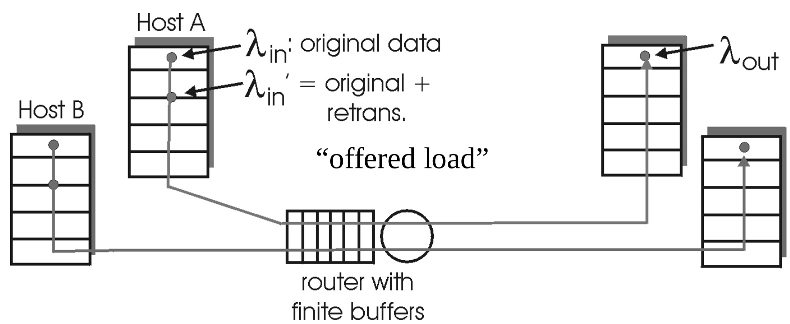




- large delays when congested
- maximum achievable throughput

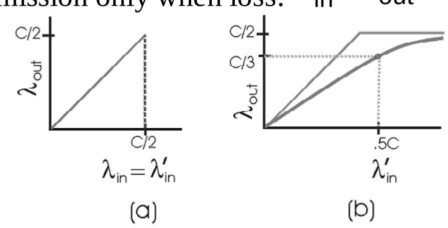
Causes/costs of congestion: scenario 2

- one router, **finite** buffers
- sender retransmission of lost packet



Causes/costs of congestion: scenario 2

- if: $\lambda_{in} = \lambda'_{in}$, (goodput)
- retransmission only when loss: $\lambda'_{in} > \lambda_{out}$



- Even worse: retransmission of delayed (not lost) packet makes larger λ'_{in} than the previous case for the same λ_{out}

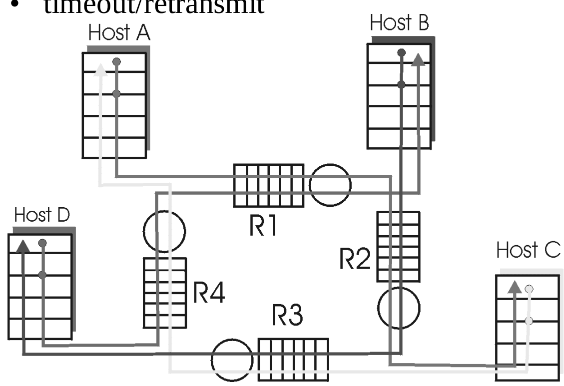
"costs" of congestion:

- more work (retrans) for given "goodput"
- unneeded retransmissions: link carries multiple copies of pkt

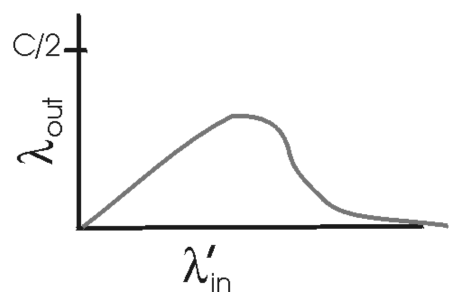
Causes/costs of congestion: scenario 3

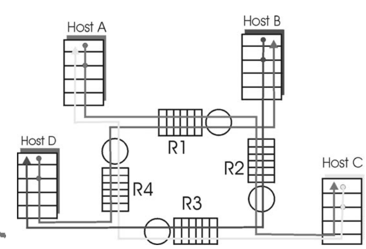
- four senders
- multihop paths
- timeout/retransmit

<u>Q:</u> what happens as λ_{in} and λ'_{in} increase ?



Causes/costs of congestion: scenario 3





Another "cost" of congestion:

- when packet dropped, any "upstream transmission capacity used for that packet was wasted!
- Throughput goes to 0 as the heavy traffic approaches infinity
- In everyone's best interest to "back off" on transmission

Approaches towards congestion control

Two broad approaches towards congestion control:

End-end congestion control:

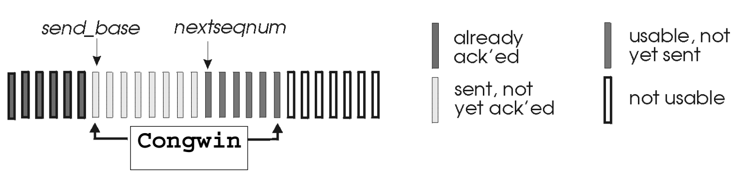
- no explicit feedback from network
- congestion inferred from end-system observed loss, delay
- approach taken by TCP

Network-assisted congestion control:

- routers provide feedback to end systems
  - single bit indicating congestion (SNA, DECbit, TCP/IP ECN, ATM)
  - explicit rate sender should send at

TCP Congestion Control

- end-end control (no network assistance)
- transmission rate limited by congestion window size, **`Congwin`**, over segments:



- w segments, each with MSS bytes sent in one RTT:

$$\boxed{\text{throughput} = \frac{w * MSS}{RTT} \text{ Bytes/sec}}$$

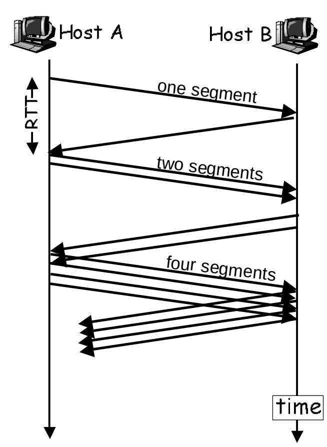
TCP congestion control:

- "probing" for usable bandwidth:
  - **ideally:** transmit as fast as possible (**`Congwin`** as large as possible) without loss
  - **Reality:**
  - *increase* **`Congwin`** until loss (congestion)
  - loss: *decrease* **`Congwin`**, then begin probing (increasing) again
- two "phases"
  - slow start
  - congestion avoidance
- important variables:
  - **`Congwin`**
  - **`threshold:`** defines threshold between two slow start phase, congestion control phase

TCP Slowstart

Slowstart algorithm

```
initialize: Congwin = 1
for (each segment ACKed)
      Congwin++
until (loss event OR
       CongWin > threshold)
```

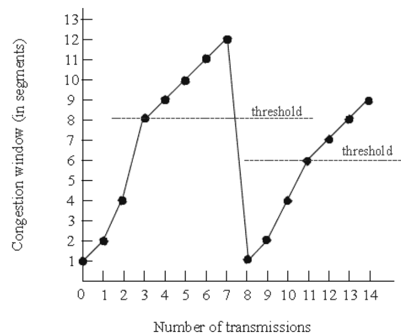


- exponential increase (per RTT) in window size (not so slow!)
- loss event: timeout (Tahoe TCP) and/or or three duplicate ACKs (Reno TCP)
- (What causes duplicate ACKs?)

TCP Congestion Avoidance

Congestion avoidance

```
/* slowstart is over       */
/* Congwin > threshold */
Until (loss event) {
  every w segments ACKed:
      Congwin++
  }
threshold = Congwin/2
Congwin = 1
perform slowstart[1]
```



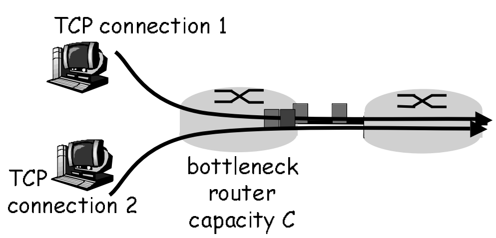
AIMD

TCP congestion avoidance:

- AIMD: *additive increase, multiplicative decrease*
  - increase window by 1 per RTT
  - decrease window by factor of 2 on loss event

TCP Fairness

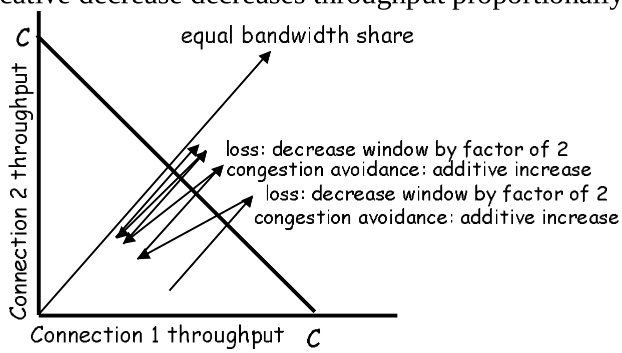
Fairness goal: if N TCP sessions share same bottleneck link, each should get 1/N of link capacity



Why is TCP fair?

Two competing sessions:

- Additive increase gives slope of 1, as throughput increases
- multiplicative decrease decreases throughput proportionally



Eventually the two connections fluctuate along equal bandwidth line

TCP vs. UDP

- When to use TCP
  - Need reliable network service
  - Want flow, congestion control
- When to use UDP
  - Don't want overhead of TCP
  - Don't want congestion control! I.e. we don't want to be "fair"
    - Multimedia apps
    - Don't want data rate throttled, but ironically this can lead to unfair transmission rate and could actually bring all traffic to a halt
    - Could also be unfair using TCP by opening multiple parallel connections (often done with web data)